

This Page Is Inserted by IFW Operations
and is not a part of the Official Record

BEST AVAILABLE IMAGES

Defective images within this document are accurate representations of the original documents submitted by the applicant.

Defects in the images may include (but are not limited to):

- BLACK BORDERS
- TEXT CUT OFF AT TOP, BOTTOM OR SIDES
- FADED TEXT
- ILLEGIBLE TEXT
- SKEWED/SLANTED IMAGES
- COLORED PHOTOS
- BLACK OR VERY BLACK AND WHITE DARK PHOTOS
- GRAY SCALE DOCUMENTS

IMAGES ARE BEST AVAILABLE COPY.

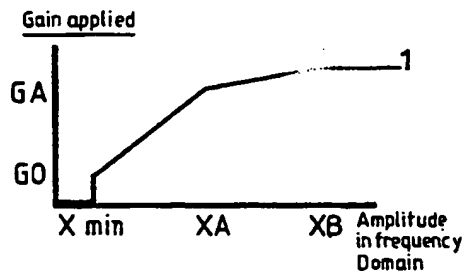
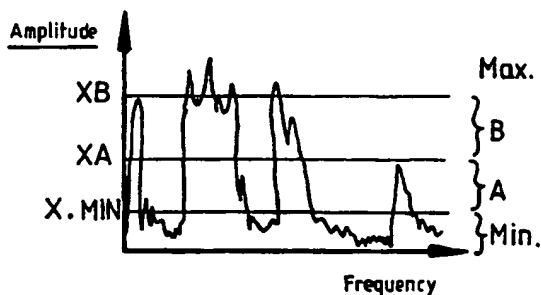
**As rescanning documents *will not* correct images,
please do not report the images to the
Image Problem Mailbox.**



INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(51) International Patent Classification ⁶ : H03H 17/02, H03G 9/00	A1	(11) International Publication Number: WO 95/25382 (43) International Publication Date: 21 September 1995 (21.09.95)
<p>(21) International Application Number: PCT/GB95/00602</p> <p>(22) International Filing Date: 17 March 1995 (17.03.95)</p> <p>(30) Priority Data: 9405211.5 17 March 1994 (17.03.94) GB</p> <p>(71) Applicant (for all designated States except US): JABRA CORPORATION [US/US]; Suite 330, 9191 Towne Centre Drive, San Diego, CA 92122 (US).</p> <p>(72) Inventor; and (75) Inventor/Applicant (for US only): DEAS, Alexander, Roger [GB/GB]; 8 Eskview Grove, Dalkeith, Edinburgh EH22 1JW (GB).</p> <p>(74) Agents: McCALLUM, William, Potter et al.; Cruikshank & Fairweather, 19 Royal Exchange Square, Glasgow G1 3AE (GB).</p>		<p>(81) Designated States: JP, NO, US, European patent (AT, BE, CH, DE, DK, ES, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE).</p> <p>Published With international search report. Before the expiration of the time limit for amending the claims and to be republished in the event of the receipt of amendments.</p>

(54) Title: NOISE CANCELLATION SYSTEM AND METHOD



(57) Abstract

A system and method for cancelling noise in an analogue signal. The analogue signal is received and converted into a digital signal using an analogue to digital converter. The digitised signal is subsequently transformed into the frequency domain using a fast Fourier transform method. Frequency components of the frequency domain signal having an amplitude less than a predetermined threshold level are subsequently attenuated, and the filtered signal is reconverted into the time domain using an inverse fast Fourier transform. The filtered time domain signal is then reconverted into an analogue signal using a digital to analogue converter.

FOR THE PURPOSES OF INFORMATION ONLY

Codes used to identify States party to the PCT on the front pages of pamphlets publishing international applications under the PCT.

AT	Austria	GB	United Kingdom	MR	Mauritania
AU	Australia	GE	Georgia	MW	Malawi
BB	Barbados	GN	Guinea	NE	Niger
BE	Belgium	GR	Greece	NL	Netherlands
BF	Burkina Faso	HU	Hungary	NO	Norway
BG	Bulgaria	IE	Ireland	NZ	New Zealand
BJ	Benin	IT	Italy	PL	Poland
BR	Brazil	JP	Japan	PT	Portugal
BY	Belarus	KE	Kenya	RO	Romania
CA	Canada	KG	Kyrgystan	RU	Russian Federation
CF	Central African Republic	KP	Democratic People's Republic of Korea	SD	Sudan
CG	Congo	KR	Republic of Korea	SE	Sweden
CH	Switzerland	KZ	Kazakhstan	SI	Slovenia
CI	Côte d'Ivoire	LI	Liechtenstein	SK	Slovakia
CM	Cameroon	LK	Sri Lanka	SN	Senegal
CN	China	LU	Luxembourg	TD	Chad
CS	Czechoslovakia	LV	Larvia	TG	Togo
CZ	Czech Republic	MC	Monaco	TJ	Tajikistan
DE	Germany	MD	Republic of Moldova	TT	Trinidad and Tobago
DK	Denmark	MG	Madagascar	UA	Ukraine
ES	Spain	ML	Mali	US	United States of America
FI	Finland	MN	Mongolia	UZ	Uzbekistan
FR	France			VN	Viet Nam
GA	Gabon				

NOISE CANCELLATION SYSTEM AND METHOD

The present invention relates to a system and method for reducing the noise in a signal.

It is often the case that transmitted electrical signals conveying information are contaminated by noise which may be manmade or which may originate from the natural environment. This noise can introduce errors and/or distortions into the information being conveyed which are particularly significant when the signal to noise ratio is low.

The problem of noise is particularly significant with modern telephone headsets which do not use a "boom" microphone but which instead have a single earpiece which contains both a microphone and earphone. Because of the relative remoteness of the microphone from the operator's mouth, extraneous background noise, for example from neighbouring operators, can be significant.

A number of different methods have been employed in the past to reduce the level of background noise in electrical signals. These methods include frequency shaping and voice switching (US 4,879,746) which attenuates any signals below a certain threshold, and conditions the signals above a certain threshold by using frequency shaping or adjusting the output signal amplitudes. However, in general these methods have proved unsuitable for "all-in-the-ear" type headsets.

It is an object of the present invention to overcome or at least mitigate the disadvantages of existing noise cancellation systems and methods.

According to a first aspect of the present invention there is provided a method of reducing noise in a signal containing an information component and a noise component and comprising converting the signal from the time domain to the frequency domain, defining at least two amplitude bands in the frequency domain and assigning each frequency component into one of the bands, attenuating the frequency components in the lower band(s) relative to those in the higher band(s), and reconverting the

-2-

resulting frequency domain signal into the time domain.

The present invention results from the realisation that when a speech signal is viewed in the frequency domain, the noise present tends to fill the space in the frequency domain which contains little or no information relating to speech. Attenuating the noise-related frequency components greatly reduces the total noise power present in the signal because the noise is wideband.

10 In a preferred embodiment of the present invention there are defined at least three, and more preferably four amplitude bands, the threshold amplitudes separating the bands being adaptively varied in dependence upon the power in the input signal. Frequency components having an amplitude within the lowest band are completely eliminated by multiplying with a gain value of zero whilst frequency components having an amplitude within the highest frequency band are preserved by multiplying them with a gain value of one. Frequency components having amplitudes within the intermediate band(s) are attenuated, or de-emphasised, by multiplying them with a gain value of between zero and one. For the intermediate bands, the gain value is determined by a gain function, a distinct gain function existing for each intermediate band where several exist, which defines a gain value which increases with component amplitude.

20 According to a second aspect of the present invention there is provided a noise reduction system comprising an input for receiving a signal containing an information component and a noise component, conversion means for converting the signal from the time domain to the frequency domain, comparator means defining at least two amplitude bands in the frequency domain and for assigning each frequency component into one of the bands, amplitude scaling means for attenuating the frequency components in the lower band(s) relative to those in the higher band(s), and reversion means for reversioning

-3-

the resulting frequency domain signal into the time domain.

According to a further aspect of the present invention there is provided a noise cancellation system comprising a unit for receiving and transmitting a signal, the unit comprising:

analogue to digital conversion means for receiving a signal which is contaminated with noise and converting the signal into a digital signal;

digital signal processing means to perform a fast Fourier transform (FFT) operation on the digital signal and to remove any spectral line which is lower in amplitude than a certain predetermined threshold value, and finally to convert the frequency domain digital signal back to a time domain signal by doing an inverse FFT operation;

digital to analogue conversion means to convert the new digital signal to an analogue signal and to transmit the new signal;

a random access read/write memory to store digital values of incoming signals and results of operations; and a non-volatile memory which stores the algorithm for the processor to perform the necessary function.

For a better understanding of the present invention and in order to show how the same may be carried into effect reference will now be made, by way of example, to the accompanying drawings in which:

Figure 1 shows schematically a noise cancellation system embodying the present invention;

Figure 2 is a flow diagram illustrating the main stages of the noise cancellation process in the system of Figure 1;

Figures 3a to 3f show signals present at various stages in the process of Figure 2 when a first noise cancellation algorithm is employed;

Figure 4 shows a set of amplitude bands into which frequency components are assigned according to an

-4-

enhanced noise cancellation algorithm; and

Figure 5 shows a gain weighting function applied to a frequency domain signal in the algorithm of Figure 2.

There will now be described with reference to
5 Figures 1 to 3 a noise cancellation unit suitable for use with an all-in-the-ear headset of the type described above. The unit is preferably arranged to plug into a modified telephone and is connected by a wire to the all-in-the-ear headset.

10 A signal processing unit 1 receives an analogue electrical signal 2 which is typically contaminated by noise. The noise may arise, for example, from the earphone which sits beside the microphone or from neighbouring persons. The input signal is sampled at a
15 rate of 8KHz and is transmitted to a flash type analogue to digital converter (ADC) 3 which converts the analogue signal into its digital equivalent. The digital signal is received by a digital signal processor (DSP) 4 which buffers the incoming signal in blocks of 64 points.

20 The software for controlling the operation of the digital signal processor is stored in a non-volatile ROM memory device 5 coupled to the DSP 4. A RAM memory device 6 is also coupled to the DSP and receives the operating code from the ROM 5 on power-up. The RAM 6
25 also acts as a buffer for the incoming 64 sample block.

Each block is pre-emphasised before further processing. Pre-emphasis is a standard technique for compensating for the average spectral shape of speech signals and involves subtracting the previous sample
30 point from the current sample point.

The current 64 point block is combined with the previous 64 point block to provide an enlarged 128 sample point frame. The enlarged frame is windowed using a standard Hanning window for the purpose of increasing the
35 frequency resolution of the subsequent transform process. The resulting windowed frame is then converted into the frequency domain using a fast Fourier transform

-5-

technique. The result of this transform is a frame in the frequency domain in which the signal is defined at a number of discrete frequencies or "bins". The frequency bins are approximately 61Hz wide.

5 At this stage a pre-boost algorithm is applied to the frame to compensate the signal for non-linearities in the response of the microphone, or other components of the system, and involves multiplying each sample point in the block by a constant defined in a look-up table. The
10 look-up table may be determined empirically.

 The DSP 4 then performs a weighting function on the signal to ease identification of the relevant part of the signal. Any spectral line which is lower in amplitude than a certain predetermined threshold value is removed
15 at this stage.

 Once the processing is complete the DSP 4 applies a post-boost algorithm to the frame. The purpose of this is similar to that of the pre-boost algorithm, i.e. to compensate for non-linearities in the signal. An inverse
20 FFT is then applied to the frame to convert it into the time domain. The time domain digital signal is then converted to an analogue signal by a digital to analogue converter DAC 7.

 The resulting signal is a time domain block having
25 128 sample points. The central 64 sample points are then extracted. In order to create an analogue output signal a de-emphasis function is applied in order to compensate for the initial pre-emphasis function. This step involves adding the present sample to the previously
30 processed sample.

 Figure 3 shows examples of typical signals at various stages of processing. Figure 3a shows a typical analogue signal input to the unit 1. Figure 3b shows the same signal after an FFT operation has been performed
35 although in reality this signal will be in digital rather than analogue form. The x-axis in Figure 3b is a measure of frequency rather than time. Figure 3c shows the FFT

-6-

signal after the pre-boost has been performed. Figure 3d shows the after effects of removal of spectral lines which were lower than a predetermined threshold. Figure 3e shows the spectrum after post-boost has been performed on the spectrum of Figure 3d. Figure 3f shows the final signal after an inverse FFT has been performed.

As an alternative to the simple noise cancellation step described above, a more complex adaptive scaling process may be employed in which, after the pre-boost step, the frequency components are partitioned and scaled according to their amplitude. The flow diagram of Figure 2 is also applicable to this modification although the details of the 'scaling' algorithm are changed.

As illustrated in Figure 4, the frequency components are assigned to one of four amplitude bands MIN, A, B, and MAX. These bands are defined by the threshold levels X_{MIN} , X_A and X_B . Each amplitude band has a corresponding gain function which is applied to the frequency components within that amplitude band and these gain functions are shown in Figure 4.

Frequency components which have an amplitude lying within the MAX band, i.e. having an amplitude greater than X_B , are assumed to correspond only to speech signals and are therefore are multiplied by a gain of one. The frequency components situated within band B are multiplied with a ramped gain such that frequency components nearer the lower end of the band are attenuated relative to frequency components near the upper end of the band. The level X_B which defines the upper end of band B is not constant and is varied depending upon the total power present in the signal, i.e. $X_B = K_B \times \text{total power}$, where K_B is a constant less than one.

Similarly, frequency components having an amplitude located within band A are multiplied by a ramped gain function. However, the ramp of band A is significantly steeper than that of band B. Again, the level X_A

-7-

defining the upper level of band A is dependent upon the total power present in the signal such that $X_A = X_B \times K_A$, where K_A is also a constant less than 1.

5 The upper level of band MIN is defined by X_{MIN} which has a fixed value. Frequency components located within this band are multiplied by a gain of zero and are thus effectively cancelled.

10 The slopes of the gain ramps for bands A and B are determined by the values of G_0 and G_A as shown in Figure 5.

15 It will be appreciated by the skilled person that various modifications may be made to the above described embodiment without departing from the scope of the present invention. For example, conversion of a time domain signal into the frequency domain may be achieved by methods other than a fast fourier transform. For example, wavelet transforms may be used or any other type of frequency transform including discrete fourier transforms (DFT).

-8-

Claims

1. A method of reducing noise in a signal containing an information component and a noise component and comprising converting the signal from the time domain to the frequency domain, defining at least two frequency amplitude bands and assigning each frequency component into one of the bands, attenuating the frequency components in the lower band(s) relative to those in the higher band(s), and reconvertng the resulting frequency domain signal into the time domain.
2. A method according to claim 1, wherein the step of attenuating the relatively low amplitude frequency components comprises multiplying the amplitude of each frequency component by a gain value, the gain value being determined from a gain function which is distinct for each amplitude band.
3. A method according to claim 2, wherein the gain function for the upper amplitude band is one and the gain function for the lower amplitude band is zero.
4. A method according to claim 2 or 3, wherein at least three bands are defined, the gain function for the or each central band being a ramp function which increases proportionally to frequency component amplitude.
5. A method according to any one of the preceding claims, wherein at least four bands are defined.
6. A method according to any one of the preceding claims and comprising the step of adaptively varying the threshold level(s) which defines the bands, in dependence upon the power of the input signal.
7. A method according to any one of the preceding claims, wherein the input signal is an analogue signal and the method comprises the step of converting the input signal into a digital signal prior to converting it into the frequency domain and the step of converting the output time domain signal into an analogue signal.
8. A method according to claim 7, wherein the time to

-9-

frequency domain conversion is carried out using a discrete Fourier transform and the frequency to time domain reversion is carried out using an inverse discrete Fourier transform.

5 9. A method according to any one of the preceding claims and comprising the step of compensating the frequency domain signal, prior to attenuating the lower frequency components, using an empirically derived frequency shaping table to compensate for signal
10 distortion.

10. A method according to any one of the preceding claims for use in removing background noise from an audio signal.

11. A noise reduction system comprising an input for
15 receiving a signal containing an information component and a noise component, conversion means for converting the signal from the time domain to the frequency domain, comparator means defining at least two frequency amplitude bands and for assigning each frequency
20 component into one of the bands, amplitude scaling means for attenuating the frequency components in the lower band(s) relative to those in the higher band(s), and reversion means for reversioning the resulting frequency domain signal into the time domain.

25 12. A telephone or telephone interface comprising a noise reduction system according to claim 11.

13. A noise cancellation system comprising a unit for receiving and transmitting a signal, the unit comprising:
analogue to digital conversion means for receiving a
30 signal which is contaminated with noise and converting the signal into a digital signal;

digital signal processing means to perform a fast Fourier transform (FFT) operation on the digital signal and to remove any spectral line which is lower in
35 amplitude than a certain predetermined threshold value, and finally to convert the frequency domain digital signal back to a time domain signal by doing an inverse

-10-

FFT operation;

digital to analogue conversion means to convert the new digital signal to an analogue signal and to transmit the new signal;

- 5 a random access read/write memory to store digital values of incoming signals and results of operations; and
a non-volatile memory which stores the algorithm for the processor to perform the necessary function.

1-3

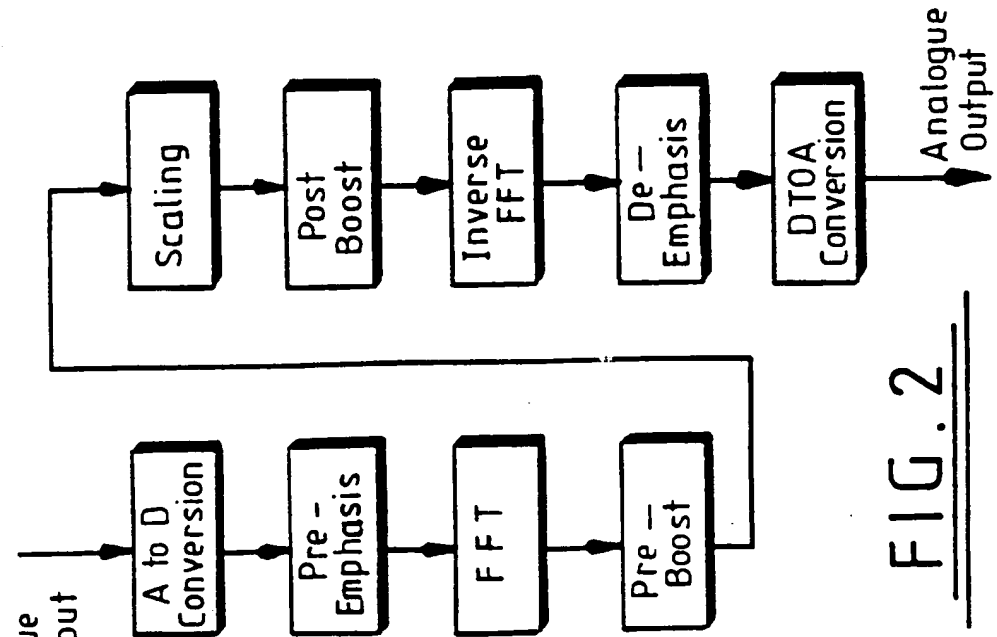


FIG. 2

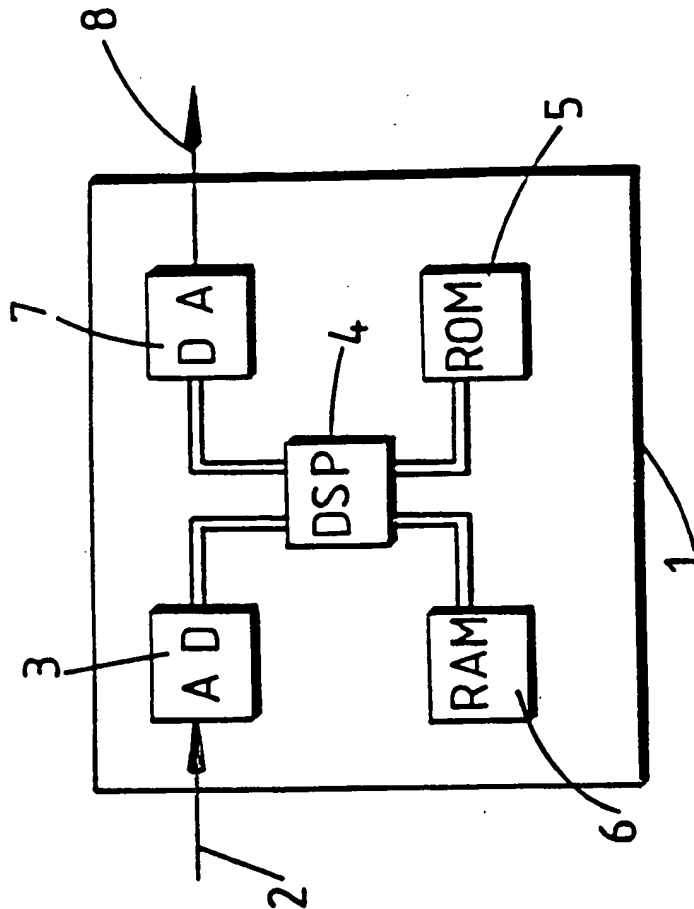
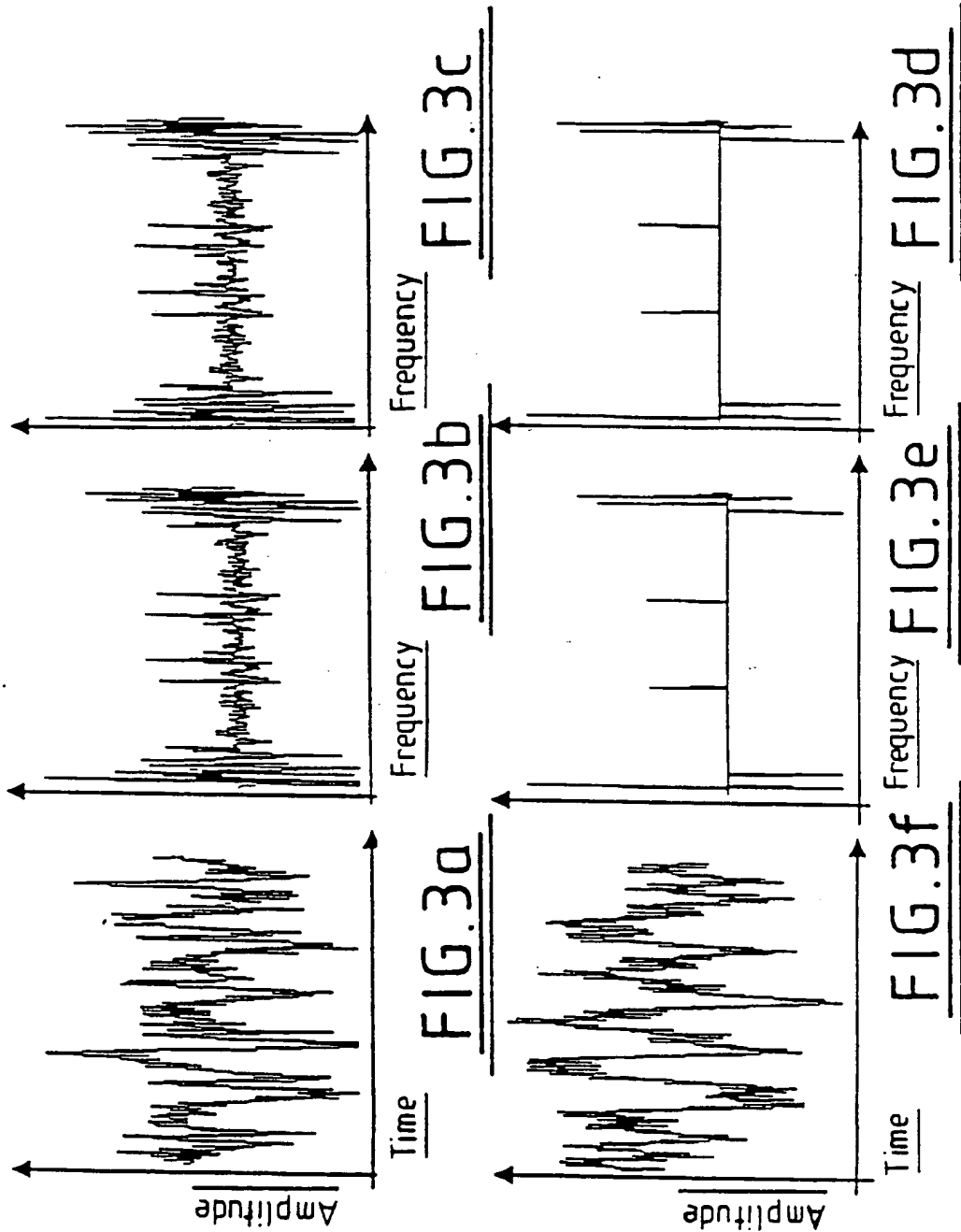


FIG. 1



3-3

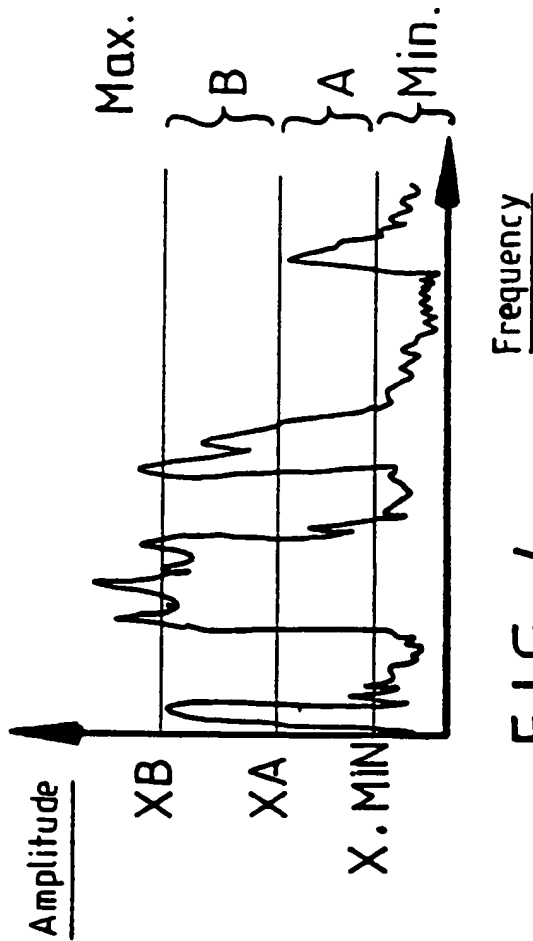


FIG. 4

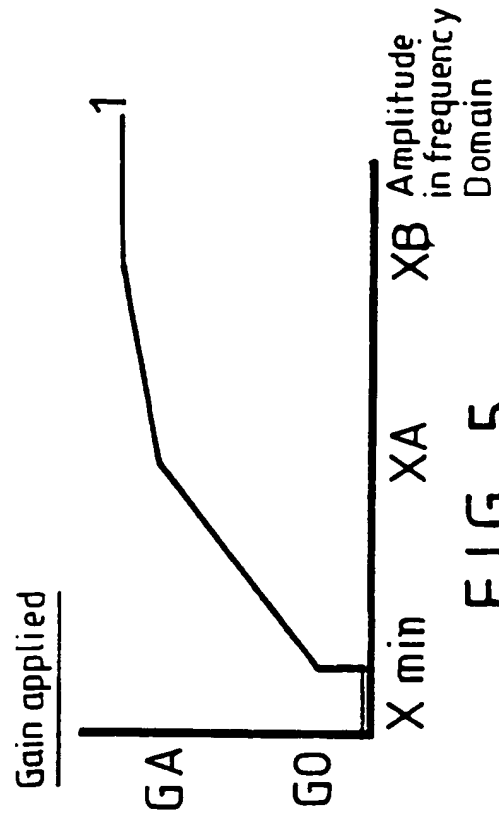


FIG. 5

INTERNATIONAL SEARCH REPORT

International Application No

PCT/GB 95/00602

A. CLASSIFICATION OF SUBJECT MATTER
 IPC 6 H03H17/02 H03G9/00

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC 6 H03H H03G

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	WO,A,89 06877 (BRITISH TELECOM) 27 July 1989 see page 2, line 4 - page 2, line 17 see page 4, line 30 - page 5, line 30 see page 14, line 19 - page 15, line 23 ---	1,7,8, 10-12
A	EP,A,0 529 158 (LOVEJOY CONTROLS) 3 March 1993 see column 2, line 50 - column 4, line 11 -----	1

☐ Further documents are listed in the continuation of box C.

☒ Patent family members are listed in annex.

* Special categories of cited documents:

- *A* document defining the general state of the art which is not considered to be of particular relevance
- *E* earlier document but published on or after the international filing date
- *L* document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)
- *O* document referring to an oral disclosure, use, exhibition or other means
- *P* document published prior to the international filing date but later than the priority date claimed

- *T* later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
- *X* document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
- *Y* document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.
- *&* document member of the same patent family

Date of the actual completion of the international search

1 August 1995

Date of mailing of the international search report

07.08.95

Name and mailing address of the ISA

European Patent Office, P.B. 5818 Patentlaan 2
 NL - 2280 HV Rijswijk
 Tel. (+31-70) 340-2040, Tx. 31 651 epo nl,
 Fax (+31-70) 340-3016

Authorized officer

Coppieters, C

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No

PCT/GB 95/00602

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
WO-A-8906877	27-07-89	CA-A- 1332626	18-10-94
		DE-D- 68913139	24-03-94
		DE-T- 68913139	23-06-94
		EP-A- 0367803	16-05-90
		GB-A- 2220330	04-01-90
		JP-T- 2503256	04-10-90
		US-A- 5133013	21-07-92
<hr/>			
EP-A-529158	03-03-93	US-A- 5046101	03-09-91
<hr/>			